AMENDMENTS TO THE CLAIMS

A complete marked-up listing of the claims, with status identifiers for each claim in parenthesis, is as follows:

1-18 (Canceled).

19. (Previously Presented) A waveform quality measurement apparatus, comprising:

an optimization circuit configured to provide a plurality of offsets of parameters of an actual signal with respect to an ideal signal;

a compensation circuit configured to compensate the actual signal with the plurality of offsets to generate a compensated actual signal;

a filtering circuit configured to filter the compensated actual signal to generate a filtered signal; and

a processor configured to modify the ideal signal to correspond to the filtered signal to generate a modified signal, and configured to determine the waveform quality measurement in accordance with the modified ideal signal and the filtered signal.

20. (Previously Presented) The apparatus of claim 19, wherein the optimization circuit is configured to provide a frequency offset, a time offset, and a phase offset.

21. (Previously Presented) The apparatus of claim 19, wherein the compensation circuit is configured to compensate the actual signal with the plurality of offsets in accordance with the following equation:

$$y(t) = x(t - \hat{\tau}_0)e^{j\left[\Delta\hat{\omega}\cdot t + \hat{\theta}_0\right]}$$

where:

$y(t)$ is the compensated actual signal;

$x(t)$ is the actual signal;

t is time;

j is an imaginary unit;

$\Delta\hat{\omega}$ is the frequency offset;

$\hat{\tau}_0$ is the time offset; and

$\hat{\theta}_0$ is the phase offset.

22. (Previously Presented) The apparatus of claim 19, wherein the filtering circuit is configured to filter the compensated signal by assigning the compensated actual signal a value that is zero in intervals to be filtered and non-zero elsewhere.

23. (Previously Presented) The apparatus of claim 22, wherein the filtering circuit is configured to filter the compensated signal by assigning the compensated actual signal a value that is non-zero over an elementary unit of the actual signal.

24. (Previously Presented) The apparatus of claim 22, wherein the filtering circuit assigns the compensated actual signal value by defining a function with a value that is zero in intervals to be filtered and non-zero elsewhere, and by multiplying the compensated actual signal by the function.

25. (Previously Presented) The apparatus of claim 24, wherein the filtering circuit defines a function with a value that is non-zero over an elementary unit of the actual signal.

26. (Previously Presented) The apparatus of claim 19, wherein the processor generates the modified ideal signal to have a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

27. (Previously Presented) The apparatus of claim 19, wherein the processor is configured to modify the ideal signal by assigning the ideal signal a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

28. (Previously Presented) The apparatus of claim 19, wherein the processor assigns the ideal signal a value_-by defining a function with a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere, and by multiplying the compensated actual signal by the function.

29. (Previously Presented) The apparatus of claim 23, wherein the processor is configured to determine the waveform quality by calculating a first overall modulation accuracy.

30. (Previously Presented) The apparatus of claim 29, wherein the processor calculates a first modulation accuracy by calculating in accordance with the following equation:

$$\rho_{\text{overall-1}} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{overall-1}$ is the first overall modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

31. (Previously Presented) The apparatus of claim 29, wherein the processor is further configured to determine the waveform quality by calculating a second overall modulation accuracy.

32. (Previously Presented) The apparatus of claim 31, wherein the processor calculates a second modulation accuracy in accordance with the following equation:

$$\rho_{\text{overall-2}} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{overall-2}$ is the second modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[(M + \frac{M}{2} + 1) \cdot (j-1) + k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[(M + \frac{M}{2} + 1) \cdot (j-1) + k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

33. (Previously Presented) The apparatus of claim 22, wherein the processor is configured to determine the waveform quality by calculating a modulation accuracy for a time division channel.

34. (Previously Presented) The apparatus of claim 33, wherein the processor calculates a modulation accuracy for a time division channel in accordance with the following equation:

$$\rho_{TDM\_channel} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^{2}}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^{2} \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^{2} \right\}}$$

where:

$\rho_{TDM\_channel}$ is the modulation accuracy for the time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

35. (Previously Presented) The apparatus of claim 22, wherein the processor is configured to determine the waveform quality measurement by calculating code domain power coefficients.

36. (Previously Presented) The apparatus of claim 35, wherein the processor calculates code domain power coefficients in accordance with the following equation:

$$\rho_{TDM\_channel,i} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{i,j,k}^{*} \right|^{2}}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{i,j,k}^{\cdot} \right|^{2} \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^{2} \right\}}, \quad i = w_1, \ldots, w_v$$

where:

$\rho_{TDM\_channel,i}$ is the code domain coefficient for a time division channel *TDM_channel* and a code channel *i*;

w1 is a first code channel for the time division channel *TDM_channel*;

wv is a last code channel for time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a kth sample in the jth elementary unit of the filtered signal; and $R'_{i,j,k} = R'_i\left[M(j-1)+k\right]$ is a kth sample in the jth elementary unit of the i-th code channel of the ideal signal.

37. (Currently Amended) A mobile station configured to measure waveform quality of a received signal using the waveform quality measurement apparatus of claim 19.

38. (Currently Amended) A base station configured to measure waveform quality of a received signal using the waveform quality measurement apparatus of claim 19.

39. (Previously Presented) A computer-readable medium including program code stored thereon, for determining a waveform quality measurement, comprising:

program code to provide a plurality of offsets of parameters of an actual signal with respect to an ideal signal;

program code to compensate the actual signal with the plurality of offsets to generate a compensated actual signal;

program code to filter the compensated actual signal to generate a filtered signal;

program code to modify the ideal signal to correspond to the filtered signal to generate a modified signal; and

program code to determine the waveform quality measurement in accordance with the modified ideal signal and the filtered signal.

40. (Previously Presented) The computer-readable medium of claim 39, wherein the program code to provide a plurality of offsets comprises program code to provide a frequency offset, a time offset, and a phase offset.

41. (Previously Presented) The computer-readable medium of claim 39, wherein the program code to compensate the actual signal with the plurality of offsets comprises program code to compensate in accordance with the following equation:

$$y(t) = x(t - \hat{\tau}_0)e^{j[\Delta\hat{\omega}\cdot t + \hat{\theta}_0]}$$

where:

$y(t)$ is the compensated actual signal;

$x(t)$ is the actual signal;

t is time;

j is an imaginary unit;

$\Delta\hat{\omega}$ is the frequency offset;

$\hat{\tau}_0$ is the time offset; and

$\hat{\theta}_0$ is the phase offset.

42. (Previously Presented) The computer-readable medium of claim 39, wherein the program code to filter comprises program code to assign the compensated actual signal a value that is zero in intervals to be filtered and non-zero elsewhere.

43. (Previously Presented) The computer-readable medium of claim 42, wherein the program

code to filter comprises program code to assign the compensated actual signal a value that is non-zero over an elementary unit of the actual signal.

44. (Previously Presented) The computer-readable medium of claim 42, wherein the program code to assign the compensated actual signal value comprises:

program code to define a function with a value that is zero in intervals to be filtered and non-zero elsewhere; and

program code to multiply the compensated actual signal by the function.

45. (Previously Presented) The computer-readable medium of claim 44, wherein the program code to define a function comprises program code to define a function with a value that is non-zero over a elementary unit of the actual signal.

46. (Previously Presented) The computer-readable medium of claim 39, wherein the program code to modify the ideal signal comprises program code to generate the modified ideal signal to have a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

47. (Previously Presented) The computer-readable medium of claim 39, wherein the program code to modify the ideal signal comprises program code to assign the ideal signal a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere.

48. (Previously Presented) The computer-readable medium of claim 47, wherein the program code to assign the ideal signal a value comprises:

program code to define a function with a value that is zero in intervals where the filtered signal has a value of zero and non-zero elsewhere; and

program code to multiply the compensated actual signal by the function.

49. (Previously Presented) The computer-readable medium of claim 43, wherein the program

code to determine the waveform quality comprises program code to calculate a first overall modulation accuracy.

50. (Previously Presented) The computer-readable medium of claim 49, wherein the program code to calculate a first modulation accuracy comprises program code to calculate in accordance with the following equation:

$$\rho_{overall\text{-}1} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^{2}}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^{2} \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^{2} \right\}}$$

where:

$\rho_{overall-1}$ is the first overall modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

51. (Previously Presented) The computer-readable medium of claim 49, further comprising program code to calculate a second overall modulation accuracy.

52. (Previously Presented) The computer-readable medium of claim 51, wherein the program code to calculate a second modulation accuracy comprises program code to calculate in accordance with the following equation:

$$\rho_{overall\text{-}2} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=\frac{M}{2}+1}^{M+\frac{M}{2}+1} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{overall-2}$ is the second modulation accuracy;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[(M + \frac{M}{2} + 1) \cdot (j - 1) + k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[(M + \frac{M}{2} + 1) \cdot (j - 1) + k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

53. (Previously Presented) The computer-readable medium of claim 42, wherein the program code to determine the waveform quality comprises program code to calculate a modulation accuracy for a time division channel.

54. (Previously Presented) The computer-readable medium of claim 53, wherein the program code to calculate a modulation accuracy for a time division channel comprises program code to calculate in accordance with the following equation:

$$\rho_{TDM\_channel} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^2 \right\}}$$

where:

$\rho_{TDM\_channel}$ is the modulation accuracy for the time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the filtered signal; and

$R_{j,k} = r[M(j-1)+k]$ is a k_{th} sample in the j_{th} elementary unit of the ideal signal.

55. (Previously Presented) The computer-readable medium of claim 42, wherein the program code to determine the waveform quality measurement comprises program code to calculate code domain power coefficients.

56. (Previously Presented) The computer-readable medium of claim 55, wherein the program code to calculate code domain power coefficients comprises program code to calculate in accordance with the following equation:

$$\rho_{TDM\_channel,i} = \frac{N \cdot \sum_{j=1}^{N} \left| \sum_{k=1}^{M} Z_{j,k} R_{i,j,k}^{*} \right|^2}{\left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| R_{i,j,k} \right|^2 \right\} \cdot \left\{ \sum_{j=1}^{N} \sum_{k=1}^{M} \left| Z_{j,k} \right|^2 \right\}}, \; i = w_1, \ldots, w_v$$

where:

$\rho_{TDM\_channel,i}$ is the code domain coefficient for a time division channel *TDM_channel* and a code channel *i*;

w1 is a first code channel for the time division channel *TDM_channel*;

wv is a last code channel for time division channel *TDM_channel*;

j is an index designating an elementary unit of signals;

N is a summation limit designating a number of elementary units;

k is an index designating a sample in the elementary unit;

M is a summation limit designating a number of samples in the elementary unit;

$Z_{j,k} = z[M(j-1)+k]$ is a kth sample in the jth elementary unit of the filtered signal; and $R'_{i,j,k} = R'_i\left[M(j-1)+k\right]$ is a kth sample in the jth elementary unit of the i-th code channel of the ideal signal.

57. (New) A method for measuring waveform quality, the method comprising:

providing a plurality of offsets of parameters of an actual signal with respect to an ideal signal;

compensating the actual signal with the plurality of offsets to generate a compensated actual signal;

filtering the compensated actual signal to generate a filtered signal; and

modifying the ideal signal to correspond to the filtered signal to generate a modified signal, and determining the waveform quality measurement in accordance with the modified ideal signal and the filtered signal.

58. (New) The method of claim 57, further comprising providing a frequency offset, a time offset, and a phase offset.

59. (New) The method of claim 57, further comprising filtering the compensated signal by assigning the compensated actual signal a value that is zero in intervals to be filtered and non-zero elsewhere.

60. (New) The method of claim 59, further comprising filtering the compensated signal by assigning the compensated actual signal a value that is non-zero over an elementary unit of the actual signal.

61. (New) The method of claim 59, further comprising assigning the compensated actual signal value by defining a function with a value that is zero in intervals to be filtered and non-zero elsewhere, and multiplying the compensated actual signal by the function.

62. (New) An apparatus for measuring waveform quality, the apparatus comprising:

means for providing a plurality of offsets of parameters of an actual signal with respect to an ideal signal;

means for compensating the actual signal with the plurality of offsets to generate a compensated actual signal;

means for filtering the compensated actual signal to generate a filtered signal; and

means for modifying the ideal signal to correspond to the filtered signal to generate a modified signal, and determining the waveform quality measurement in accordance with the modified ideal signal and the filtered signal.

63. (New) The apparatus of claim 62, further comprising means for providing a frequency offset, a time offset, and a phase offset.

64. (New) The apparatus of claim 62, further comprising means for filtering the compensated signal by assigning the compensated actual signal a value that is zero in intervals to be filtered and non-zero elsewhere.

65. (New) The apparatus of claim 62, further comprising means for filtering the compensated signal by assigning the compensated actual signal a value that is non-zero over an elementary unit of the actual signal.

66. (New) The apparatus of claim 62, further comprising means for assigning the compensated actual signal value by defining a function with a value that is zero in intervals to be filtered and non-zero elsewhere, and multiplying the compensated actual signal by the function.